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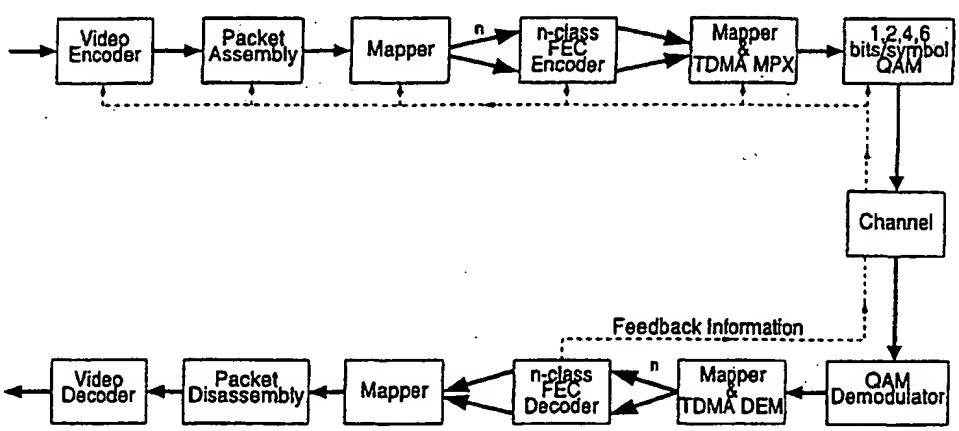

(57) Abstract

The performance benefits of burst-by-burst adaptive modulation are studied, employing a higher-order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality – such as video or audio quality – versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality – such as video or audio quality – by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors – or even hilly terrain – propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

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Title of the Invention

Burst-by-burst Adaptive Single-carrier Data Transmission

1 Background of the Invention

The invention relates to data transmission, more specifically to transmission in packets or bursts.

In contrast to the *burst-by-burst reconfigurable wideband* multimedia transceivers described in this document, the term *statically reconfigurable* found in this context in the literature refers to multimedia transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base station's control, invoking for example in the central cell region - where benign channel conditions prevail - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Modulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video quality.

The philosophy of the fixed, but programable-rate proprietary video codecs and statically reconfigurable multi-mode video transceivers presented by Streit et al in References [1]-[4] was that irrespective of the video motion activity experienced, the specially designed video codecs generated a constant number of bits per video frame. For example, for videophony over the second-generation Global System of Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the highest possible error resilience and another, aiming for the highest possible compression ratio. This fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems, such as the GSM.

The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs, such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263 codec, where the time-variant video motion activity and the variable-length coding techniques employed

result in a time-variant bitrate fluctuation and a near-constant perceptual video quality. This time-variant bitrate fluctuation can be mitigated by employing adaptive feed-back controlled buffering, which potentially increases the latency or delay of the codec and hence it is often objectionable for example in interactive videophony. The schemes presented by Streit et al in References [1]-[4] result in slightly variable video quality at a constant bitrate, while refraining from employing buffering, which again, would result in latency in interactive videophony. A range of techniques, which can be invoked, in order to render the family of variable-length coded, highly bandwidth-efficient, but potentially error-sensitive class of standard video codecs, such as the H.263 arrangement, amenable to error-resilient, low-latency interactive wireless multimode videophony was summarised in [5]. The adaptive video rate control and packetisation algorithm of [5] generates the required number of bits for the burst-by-burst adaptive transceiver, depending the on the capacity of the current packet, as determined by the current modem mode. Further error-resilient H.263-based schemes were contrived for example by Färber, Steinbach and Girod at Erlangen University [6], while Sadka, Eryurtlu and Kondoz [7] from Surrey University proposed a range of improvements to the H.263 scheme. Following the above portrayal of the prior art in both video compression and statically reconfigurable narroband modulation, let us now consider the philosophy of wideband burst-by-burst adaptive quadrature amplitude modulation (AQAM) in more depth.

In burst-by-burst adaptive modulation a higher-order modulation scheme is invoked, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, a more robust lower order modulation scheme is employed, when the channel exhibits inferior channel quality, in order to improve the mean Bit Error Ratio (BER) performance. A practical scenario, where adaptive modulation can be applied is, when a reliable, low-delay feedback path is created between the transmitter and receiver, for example by superimposing the estimated channel quality perceived by the receiver on the reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its modem mode according to this perceived channel quality.

Recent developments in adaptive modulation over a narrow-band channel environment have been pioneered by Webb and Steele [9], where the modulation adaptation was utilized in a Digital European Cordless Telephone - like (DECT) system. The concept of variable rate adaptive modulation was also advanced by Sampei *et al* [12, 17], showing promising advantages, when compared to fixed modulation in terms of spectral efficiency, BER performance and robustness against channel delay spread. In another paper, the numerical upper bound performance of adaptive modulation in a slow Rayleigh flat-fading channel was evaluated by Torrance *et al*[10] and subsequently, the optimization of the switching threshold levels using Powell minimization was used in order to achieve a targeted performance [11, 18]. In addition, adaptive modulation was also studied in conjunction with channel coding and power control

techniques by Matsuoka *et al* [12] as well as Goldsmith *et al.*[13]-[15].

In the narrow-band channel environment, the quality of the channel was determined by the short term Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose the appropriate modulation mode for the transmitter, based on a list of switching threshold levels, l_n [9, 10]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality of the channel, where the existence of multi-path components produces not only power attenuation of the transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to estimate the wideband channel quality in order to choose the appropriate modulation scheme.

2 Summary of the Invention

Particular and preferred aspects of the invention are set out in the accompanying independent and dependent claims. Features of the dependent claims may be combined with those of the independent claims as appropriate and in combinations other than those explicitly set out in the claims.

The performance benefits of burst-by-burst adaptive modulation are described, employing a higher-order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality - such as video or audio quality - versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This can achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

3 Brief Description of the Drawings

For a better understanding of the invention and to show how the same may be carried into effect reference is now made by way of example to the accompanying drawings, in which:

List of Figures

4 Detailed Description

4.1 General Introduction to Adaptive Modem Mode Signalling Scenarios

AQAM transmission parameter adaptation is an action of the transmitter in response to time–varying channel conditions. It is only suitable for duplex communication between two stations, since the transmission parameter adaptation relies on some form of channel estimation and signalling. In order to efficiently react to the changes in channel quality, the following steps have to be taken:

- *Channel quality estimation:* In order to appropriately select the transmission parameters to be employed for the next transmission, a reliable prediction of the channel quality during the next active transmit timeslot is necessary.
- *Choice of the appropriate parameters for the next transmission:* Based on the prediction of the expected channel conditions during the next timeslot, the transmitter has to select the appropriate modulation schemes for the subcarriers.
- *Signalling or blind detection of the employed parameters:* The receiver has to be informed, as to which set of demodulator parameters to employ for the received packet. This information can either be conveyed within the packet, at the cost of loss of useful data bandwidth, or the receiver can attempt to estimate the parameters employed at the transmitter by means of blind detection mechanisms.

Depending on the channel characteristics, these operations can be performed at either of the duplex stations, as shown in Figure 1. If the channel is reciprocal, then the channel quality estimation for each link can be extracted from the reverse link, and we refer to this regime as open–loop adaptation. In this case, the transmitter needs to communicate the transmission parameter set to the receiver (Figure 1(a)), or the receiver can attempt blind detection of the transmission parameters employed (Figure 1(c)).
If the channel is not reciprocal, then the channel quality estimation has to be performed at the receiver of the link. In this case, the channel quality measure or the set of requested transmission parameters is communicated to the transmitter in the reverse link (Figure 1(b)). This mode is referred to as closed–loop adaptation.

4.2 A Specific Embodiment of a Video Transceiver

The schematic of the whole system is depicted in Figure 2. In the described system the wideband channel-induced degradation is combated not only by the employment of adaptive modulation but also by equalization, where the equalization process will eliminate most of the intersymbol interference based on a

Channel Impulse Response (CIR) estimate derived using the channel sounding midamble and consequently, the signal to noise and residual interference ratio at the output of the equalizer is calculated. We note, however that the above adaptive methodology can also be extended to employing burst-by-burst adaptive channel coding associated with different-strength error correction codecs in the different transceiver modes of operation.

4.3 Channel quality metrics

The most reliable channel quality estimate is the bit error rate (BER), since it reflects the channel quality, irrespective of the source or the nature of the quality degradation.

Firstly, the BER can be estimated with a certain granularity or accuracy, provided that the system entails a channel decoder or - synonymously - Forward Error Correction (FEC) decoder employing algebraic decoding.

Secondly, if the system contains a soft-in-soft-out (SISO) channel decoder, the BER can be estimated with the aid of the Logarithmic Likelihood Ratio (LLR), evaluated either at the input or the output of the channel decoder. A particularly attractive way of invoking LLRs is employing powerful turbo codecs, which provide a reliable indication of the confidence associated with a particular bit decision in the context of LLRs. The LLR is defined as the ratio of the probabilities of a specific bit being binary zero or one. Again, this measure can be evaluated at both the input and the output of the turbo channel codecs and both of them can be used for channel quality estimation.

Thirdly, in the event that no channel encoder / decoder (codec) is used in the system, the channel quality expressed in terms of the BER can be estimated with the aid of the mean-squared error (MSE) at the output of the channel equaliser or the closely related metric, the Pseudo-Signal-to-noise-ratio (Pseudo-SNR). The MSE or pseudo-SNR at the output of the channel equaliser have the important advantage that they are capable of quantifying the severity of the inter-symbol-interference (ISI) and/or Co-channel Interference experienced, in other words quantifying the Signal to Interference plus Noise Ratio (SINR).

4.3.1 Pseudo-SNR Embodiment

A specific embodiment based on the above-mentioned pseudo-SNR is now described in more depth. Employing the pseudo-SNR has the advantage that it is generally applicable, regardless of whether or not a channel codec is present.

We found that the residual channel-induced inter-symbol-interference (ISI) at the output of the decision feedback equaliser (DFE) is near-Gaussian distributed and that if there are no decision feedback errors,

the pseudo-SNR at the output of the DFE, γ_{dfe} can be calculated as [8]:

$$\gamma_{dfe} = \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power} + \text{Effective Noise Power}} = \frac{E\left[|S_k \sum_{m=0}^{N_f-1} C_m h_m|^2\right]}{\sum_{q=-(N_f-1)}^{-1} E\left[|f_q S_{k-q}|^2\right] + N_o \sum_{m=0}^{N_f-1} |C_m|^2}. \quad (1)$$

where C_m and h_m denotes the DFE's feed-forward coefficients and the channel impulse response, respectively. The transmitted signal and the noise spectral density is represented by S_k and N_o. Lastly, the number of DFE feed-forward coefficients is denoted by N_f. By utilizing the pseudo-SNR at the output of the equalizer, we are ensuring that the system performance is optimised by employing equalization and adaptive quadrature amplitude modulation (AQAM) in a wideband environment according to the following switching regime:

$$\text{Modulation Mode} = \begin{cases} BPSK & \text{if } \gamma_{DFE} < f_1 \\ 4QAM & \text{if } f_1 < \gamma_{DFE} < f_2 \\ 16QAM & \text{if } f_2 < \gamma_{DFE} < f_3 \\ 64QAM & \text{if } \gamma_{DFE} > f_3, \end{cases} \quad (2)$$

where $f_n, n = 1 \ldots 3$ are the pseudo-SNR thresholds levels, which are set according to the system's integrity requirements.

In contrast to the narrowband, statically reconfigured multimode systems of [1]-[4] constituting the state-of-the-art, the present embodiment invokes wideband, near-instantaneously reconfigured or burst-by-burst adaptive channel-equalised modulation, in order to achieve the best possible multimedia source-signal representation quality - for example video quality - when transmitting over arbitrarily time-variant channels, exhibiting short-term and/or long-term channel quality variations. These variations can be due to the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high video quality modes are employed.

It is an important element of the system that when the binary BCH channel codes or FEC codes protecting the video stream are overwhelmed by the plethora of transmission errors, the embodiment refrains from decoding the video packet in order to prevent error propagation through the reconstructed frame buffer [5]. Instead, these corrupted packets are dropped and the reconstructed frame buffer will not be updated, until the next packet replenishing the specific video frame area arrives. The associated video

Parameter	Value
Carrier Frequency	1.9GHz
Vehicular Speed	30mph
Doppler frequency	85Hz
Normalised Doppler frequency	3.27×10^{-5}
Channel type	COST 207 Typical Urban (see Figure 3)
Number of paths in channel	4
Data modulation	Adaptive QAM (BPSK, 4-QAM, 16-QAM, 64-QAM)
Receiver type	Decision Feedback Equalizer Number of Forward Filter Taps = 35 Number of Backward Filter Taps = 7

Table 1: Modulation and channel parameters

performance degradation is fairly minor for packet dropping or frame error rates (FER) below about 5%. These packet dropping events are signalled to the remote decoder by superimposing a strongly protected one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [5]. In the embodiment we also invoked the adaptive rate control and packetisation algorithm of [5], supporting constant Baud-rate operation.

As a specific example of the burst-by-burst adaptive nultimedia system we used 176x144 pixel so-called QCIF-resolution, 30 frames/s video sequences encoded at bitrates resulting in high perceptual video quality, in order to demonstrate the performance advantages of the video transceiver. Table 1 shows the modulation- and channel parameters employed, noting again that the associated principles are applicable in the context of a whole range of other system parameters. The COST207 four-path typical urban (TU) channel model was used in quantifying the associated system performance and its impulse response is portrayed in Figure 3. As an example, we used the Pan-European FRAMES proposal as the basis for our wideband transmission system, the frame structure of which is shown in Figure 4. Employing the FRAMES Mode A1 (FMA1) so-called non-spread data burst mode required a system bandwidth of 3.9MHz, when assuming a modulation excess bandwidth of 50%. A range of other system parameters are shown in Table 2.

The specific example of the video transceiver - which is used to demonstrate the advantages of the system concept - is based on the H.263 video codec. The video coded bitstream was protected by binary Bose-

Features	Value
Multiple access	TDMA
No. of Slots/Frame	16
TDMA frame length	4.615ms
TDMA slot length	288μs
Data Symbols/TDMA slot	684
User Data Symbol Rate (KBd)	148.2
System Data Symbol Rate (MBd)	2.37
Symbols/TDMA slot	750
User Symbol Rate (KBd)	162.5
System Symbol Rate (MBd)	2.6
System Bandwidth (MHz)	3.9
Eff. User Bandwidth (kHz)	244

Table 2: Generic system features of the reconfigurable multi-mode video transceiver, using the non-spread data burst mode of the FRAMES proposal shown in Figure 4.

Chaudhuri-Hocquenghem (BCH) coding combined with an intelligent burst-by-burst adaptive wideband multi-mode Quadrature Amplitude Modulation (QAM) modem, which can be configured either under network control or under transceiver control to operate as a 1, 2, 4 and 6 bits/symbol scheme, while maintaining a constant signalling rate. This allowed us to support an increased throughput expressed in terms of the average number of bits per symbol, when the instantaneous channel quality was high, leading ultimately to an increased video quality in a constant bandwidth.

The transmitted bitrate for all four modes of operation is shown in Table 3. The unprotected bitrate before approximately half-rate BCH coding is also shown in Table 3. The actual useful bitrate available for video is slightly less, than the unprotected bitrate due to the required strongly protected packet acknowledgement information and packetisation information. The effective video bitrate is also shown in Table 3.

4.4 Burst-by-Burst Adaptive Videophone Performance

The described burst-by-burst adaptive modem maximizes the system capacity available by using the most appropriate modulation mode for the current instantaneous channel conditions. We found that the pseudo-SNR at the output of the channel equaliser was an adequate channel quality measure in our burst-

Features	Multi-rate System			
Mode	BPSK	4QAM	16QAM	64QAM
Bits/Symbol	1	2	4	6
FEC	Near Half-rate BCH			
Transmission bitrate (kbit/s)	148.2	296.4	592.8	889.3
Unprotected bitrate (kbit/s)	75.8	151.7	303.4	456.1
Effective Video-rate (kbit/s)	67.0	141.7	292.1	446.4
Video fr. rate (Hz)	30			

Table 3: Operational-mode specific transceiver parameters

by-burst adaptive wide-band modem. Figure 5 demonstrates how the burst-by-burst adaptive modem changes its modulation modes every transmission burst, ie every 4.615 ms, based on the fluctuating pseudo-SNR. The right-hand-side vertical axis indicates the associated number of bits per symbol.

By changing to more robust modulation schemes automatically, when the channel quality is reduced allows the packet loss ratio, or synonymously, the FER, to be reduced, which results in increased perceived video quality. In order to judge the benefits of burst-by-burst adaptive modulation we considered two scenarios. In the first scheme the adaptive modem always chose the perfectly estimated AQAM modulation mode, in order to provide a maximum upper bound performance. In the second scenario the modulation mode was based upon the perfectly estimated AQAM modulation mode for the previous burst, which corresponded to a delay of one Time Division Multiple Access (TDMA) frame duration of 4.615ms. This second scenario represents a practical burst-by-burst adaptive modem, where the one-frame channel quality estimation latency is due to superimposing the receiver's perceived channel quality on a reverse-direction packet, for informing the transmitter concerning the best mode to be used.

The probability of the adaptive modem using each modulation mode for a particular average channel SNR is portrayed in Figure 6 in terms of the associated modem mode probability density functions (PDFs). It can be seen at high channel SNRs that the modem mainly uses the 64QAM modulation mode, while at low channel SNRs the BPSK mode is the most prevalent one.

Figure 7 shows the transmission FER (or packet loss ratio) versus channel SNR for the 1, 2, 4 and 6 bit/symbol fixed modulation schemes, as well as for the ideal and for the one-frame delayed realistic

scenarios using the burst-by-burst adaptive QAM (AQAM) modem. In the ideal - ie zero-delay - AQAM scenario, where the modulation mode estimation is assumed to be available instantaneously, the transmission FER is zero at high channel SNRs even though 64QAM is used predominantly, while at low SNRs it exhibits a similar FER behaviour to fixed BPSK modulation, since this is the most often used mode. More explicitly, at high SNRs the adaptive modem chooses the most suitable AQAM mode and hence no packets are lost. However, at low SNRs the adaptive modem opts for using BPSK, even when the channel is so hostile that the packets are lost. Hence the BPSK and ideal - ie zero-delay - AQAM results are very similar at low channel SNRs. However, when the modulation mode estimation is delayed by one TDMA frame - representing a realistic, practical AQAM modem - then the transmission FER is no longer zero at high channel SNRs, since the delay results in a non-optimum modulation mode selection, which can result in the corresponding video packet being lost. At high channel SNRs the FER of the realistic, one-frame delay AQAM modem is similar to that of the fixed 64QAM modem mode. By contrast, at low channel SNRs its FER performance is similar to that of the fixed BPSK modem mode. However, at medium channel SNRs the transmission FER is almost constant at about 3% for the realistic AQAM modem. This is more clearly demonstrated on a logarithmic scale in Figure 8.

Explicitly, the ideal and realistic AQAM modems are compared to a fixed modulation based, statically re-configured multi-mode system with switching at 5% transmission FER in Figure 8. The statically reconfigured modem was invoked here as a benchmarker, in order to indicate, how a system would perform, which cannot act on the basis of the near-instantaneously varying channel quality. As it can be infirred from Figure 8, such a statically reconfigured transceiver switches its mode of operation from a lower-order modem mode, such as for example BPSK to a higher-order mode, such as 4QAM, when the channel quality has improved sufficiently for the 4QAM mode's FER to become lower than 5 % upon reconfiguring the transceiver in this 4QAM mode. Again - as seen in Figure 7 earlier on a non-logarithmic scale - Figure 8 clearly shows that the ideal AQAM modem has a similar FER performance to the fixed rate BPSK modem. Additionally, it indicates that the realistic AQAM modem has a similar FER performance to the BPSK modem at low SNRs, yielding a near-constant 3% FER at medium SNRs and a FER similar to that of the fixed 64QAM modem at high channel SNRs.

A comparison of the effective video bitrate versus channel SNR is shown in Figure 9 for the four fixed modulation schemes, and the ideal and realistic AQAM modems. The effective video bitrate is the average bitrate provided by all the successfully transmitted video packets. It should be noted that the realistic AQAM modem has a slightly lower throughput, since sometimes the incorrect modulation mode is chosen, which may result in packet loss. At very low channel SNRs the throughput bitrate converges to that of the fixed BPSK mode, since the AQAM modem is almost always in the BPSK mode at these

SNRs, as demonstrated in Figure 6.

Having shown the effect of the burst-by-burst adaptive modem on the transmission FER and effective bitrate, let us now demonstrate these effects on the decoded video quality, measured in terms of the Peak Signal-to-Noise Ratio (PSNR). Figure 10 shows the decoded video quality in terms of PSNR versus channel SNR for both the ideal and realistic adaptive modem, and for the four modes of the statically configured multi-mode system. It can be seen that - as expected - the ideal adaptive modem, which always selects the perfect modulation modes, has a better or similar video quality for the whole range of channel SNRs. In the statically configured multi-mode scheme the video quality degrades, when the system switches from a higher-order to a lower-order modulation mode. The ideal adaptive modem however smoothes out this sudden loss of video quality, as the channel degrades. The non-ideal adaptive modem has a slightly lower video quality performance, than the ideal adaptive modem, especially at medium SNRs, since it sometimes selects the incorrect modulation mode due to the estimation delay. This can inflict video packet loss and/or a reduction of the effective video bitrate, which in turn reduces the video quality.

The difference between the ideal burst-by-adaptive modem, using ideal channel estimation and the non-ideal, realistic burst-by-burst adaptive modem, employing a non-ideal delayed channel estimation can be seen more clearly in Figure 11 for a range of video sequences. Observe that at high and low channel SNRs the video quality performance is similar for the ideal and non-ideal adaptive modems. This is because the channel estimation delay has little effect, since at low or high channel SNRs the adaptive modem is in either BPSK or 64QAM mode for the majority of the time. More explicitly, the channel quality of a transmission frame is almost always the same as that of the next, and hence the delay has little effect at low and high SNRs.

The video quality versus channel quality trade-offs can be more explicitly observed in Figure 12. This figure portrays the decoded video quality in PSNR versus the packet loss ratio or transmission FER. The ideal and practical adaptive modem performance is plotted against that of the four fixed modulation schemes in the figure. It can been seen that the adaptive modems' video quality degrades from that achieved by the error free 64QAM modem towards the BPSK modem performance as the packet loss ratio increases. The practical adaptive modems' near constant FER performance of 3% at medium SNRs can be clearly seen in the figure, which is associated with the reduced PSNRs of the various modem modes, while having only minor channel error-induced impairments.

	BPSK	4QAM	16QAM	64QAM
Standard	<10dB	≥ 10dB	≥ 18dB	≥ 24dB
Conservative	<13dB	≥ 13dB	≥ 20dB	≥ 26dB
Aggressive	<9dB	≥ 9dB	≥ 17dB	≥ 23dB

Table 4: SINR estimate at output of the equaliser required for each modulation mode in Burst-by-Burst Adaptive modem, ie. switching thresholds

4.5 Switching Thresholds

The burst-by-burst adaptive modem changes its modulation modes based on the fluctuating channel conditions expressed in terms of the SNR at the equaliser's output. The set of switching thresholds used in all the previous graphs is the "Standard" set shown in Table 4, which was determined on the basis of the required channel SINR for maintaining the specific target video FER.

In order to investigate the effect of different sets of switching thresholds, we defined two new sets of thresholds, a more conservative set, and a more aggressive set, employing less robust, but more bandwidth-efficient modem modes at lower SNRs. The more conservative switching thresholds reduced the transmission FER at the expense of a lower effective video bitrate. The more aggressive set of thresholds increased the effective video bitrate at the expense of a higher transmission FER.

The transmission FER performance of the realistic burst-by-burst adaptive modem, which has a one TDMA frame delay between channel quality estimation and mode switching is shown in Figure 13 for the three sets of switching thresholds of Table 4. It can be seen that the more conservative switching thresholds reduce the transmission FER from about 3% to about 1% for medium channel SNRs. The more aggressive switching thresholds increase the transmission FER from about 3% to 4-5%. However, since FERs below 5% are not objectionable in video quality terms, this FER increase is an acceptable compromise for a higher effective video bitrate. The effective video bitrate for the realistic adaptive modem with the three sets of switching thresholds is shown in Figure 14. The more conservative set of switching thresholds reduces the effective video bitrate but also reduces the transmission FER. The aggressive switching thresholds, increase the effective video bitrate, but also increase the transmission FER. Therefore the optimal switching thresholds should be set such that the transmission FER is deemed acceptable is the range of channel SNRs considered.

5 Summary

The above-described burst-by-burst adaptive multimedia transceiver concept exhibits substantial advantages in comparison to conventional fixed-mode or statically reconfigurable transceivers, which was substantiated in the context of a specific embodiment of the advocated system concept, namely with the aid of a burst-by-burst adaptive video transceiver.

Specifically, the main advantage of the described burst-by-burst adaptive transceiver technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video, speech or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

The described system embodiment has the following features:

1. A reliable instantaneous channel quality metric is employed, in order to appropriately configure the AQAM modem for maintaining the required target BER and the associated source signal representation quality. The range of potential channel quality metrics entails the pseudo-SNR, SINR, BER and its LLR-based channel estimates.

2. The perceived channel quality determines the number of bits that can be transmitted in a given transmitted packet or burst, which in turn predetermines the number of bits to be generated by the associated multimedia source codec, such as for example the associated video, audio, speech or handwriting codec. Hence the multimedia source codec has to be capable of adjusting the number of bits generated under the instruction of the burst-by-burst adaptive transceiver.

3. The transmitter mode requested by the receiver, in order to achieve the target performance has to be signalled by the receiver to the remote transmitter. Another scenario is, where the uplink and downlink channel quality is sufficiently similar for allowing the receiver to judge, what transmission mode the associated transmitter should use, in order for its transmitted signal to maintain the required transmission integrity. Lastly, the mode of operation used by the transmitter can also be detected using blind detection techniques, for example in conjunction with the associated channel

decoder.

In the studied example of the system embodiment we have characterised a wideband burst-by-burst adaptive multimedia transceiver which employed the pseudo-SNR at the output of the channel equaliser as the quality measure for controlling the AQAM modem modes. Whilst in reference [16] the throughput upper-bound of such an AQAM modem was analysed, in this document a practical multimedia transceiver concept was described and the achievable performance gains due to employing the described wideband burst-by-burst adaptive modem were quantified. An adaptive packetiser was used in conjunction with the adaptive modem, which continually adjusted the video codec's target bitrate, in order to exploit the instantaneous bitrate provided by the adaptive modem.

In the example the delay between the channel estimation and modulation mode switching was shown to have a considerable effect on the performance achieved by the adaptive modem. This performance penalty can be mitigated by reducing the modem mode switching latency, for example by employing adjacent slots for the uplink and downlink of of a TDD system. However, at lower vehicular speeds the effects of AQAM mode switching latency are less crucial and the practical adaptive modem can achieve a performance that is close to that of the ideal adaptive modem exhibiting no switching latency, that we used as an upper bound benchmarker. We have also demonstrated, how the transmission FER performance is affected by changing the switching thresholds. Therefore the system can be tuned to the required FER performance using appropriate switching thresholds.

It will be appreciated that although a particular embodiment of the invention has been described, many modifications / additions and / or substitutions may be made within the spirit and scope of the present invention

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CLAIMS

1. A receiver unit comprising:

a burst-by-burst adaptive equalizer having an input for receiving data bursts from a communication channel, each burst containing a number of bits per symbol;

a computational unit for computing a received signal quality metric related to a bit error rate experienced during transmission over the communication channel;

an output for relaying the signal quality metric, conveying signal quality as perceived by the receiver unit, for use in determining a configuration for subsequent transmission bursts.

2. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from an interference parameter.

3. A receiver unit according to claim 2, wherein the signal quality metric is evaluated using channel impulse response estimates derived from a training sequence embedded in each transmission burst.

4. A receiver unit according to claim 2, wherein the interference parameter is a measure of co-channel interference.

5. A receiver unit according to claim 2, wherein the interference parameter is a measure of inter-symbol interference.

6. A receiver unit according to claim 1, wherein the signal quality metric is evaluated according to the formula:

$$\gamma_{dfe} = \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power + Effective Noise Power}} = \frac{E\left[\left|S_k \sum_{m=0}^{N_f-1} C_m h_m\right|^2\right]}{\sum_{q=-(N_f-1)}^{-1} E\left[\left|f_q S_{k-q}\right|^2\right] + N_o \sum_{m=0}^{N_f-1} \left|C_m\right|^2}.$$

where γ_{dfe} is pseudo-SNR at the output of the equalizer, C_m and H_m denote feed-forward coefficients and the channel impulse response respectively, S_k and N_0 are transmitted signal and noise spectral density respectively, and N_f is the number of feed-forward coefficients.

7. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from the bit error rate.

8. A receiver unit according to claim 7, wherein the bit error rate is estimated by an algebraic channel decoder.

9. A receiver unit according to claim 7, further comprising a channel decoder arranged to receive the data bursts from the adaptive equalizer, and wherein the bit error rate is estimated from a calculation of a logarithmic likelihood ratio, thereby to provide a reliable estimator of all possible channel impairments.

10. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the input of the channel decoder.

11. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the output of the channel decoder.

12. A receiver unit according to any one of the preceding claims, wherein the configuration defines the number of bits per symbol in each transmission burst, which is varied according to the signal quality metric computed from a previous transmission burst, as supplied by the output.

13. A receiver unit according to any one of the preceding claims, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, through the communication channel for configuration of the remote transmitter for subsequent transmission bursts, thereby to provide closed-loop feedback.

14. A receiver unit according to any one of claims 1 to 12, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, to a transmitter unit local to the receiver unit for configuration of the local transmitter unit for subsequent transmission bursts to a remote receiver unit associated with the remote transmitter unit, thereby to provide open-loop feedback.

15. A receiver unit according to any one of claims 1 to 12, wherein the signal quality metric is internally used in a blind detection scheme to reconfigure the receiver unit for decoding subsequent transmission bursts.

16. A system comprising a receiver unit according to any one of claims 1 to 12 in combination with a transmitter unit, wherein the transmitter unit has an input connected to the output of the receiver unit for receiving the signal quality metric, the transmitter unit having a configuration that is responsive to the signal quality metric.

17. A system according to claim 16, wherein the transmitter unit comprises an interactive multimedia encoder having a configuration that is responsive to the signal quality metric.

18. A system according to claim 16 or 17, wherein the transmitter unit comprises a modem having a configuration that is responsive to the signal quality metric.

19. A system according to claim 16, 17 or 18, wherein the transmitter unit comprises a channel encoder having a configuration that is responsive to the signal quality metric.

20. A system according to any one of claims 16 to 19, wherein the transmitter unit and receiver unit form a transceiver unit.

21. A system according to any one of claims 16 to 19, wherein the transmitter unit and receiver unit are remote from each other and form respective parts of separate transceiver units.

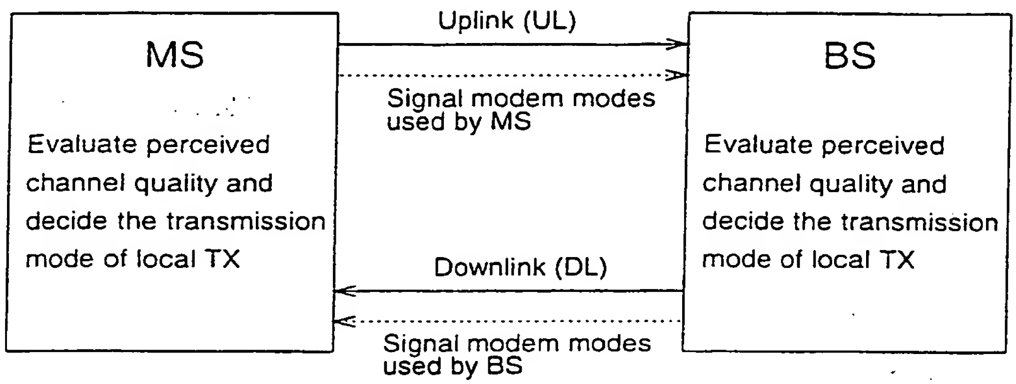


Figure 1(a)

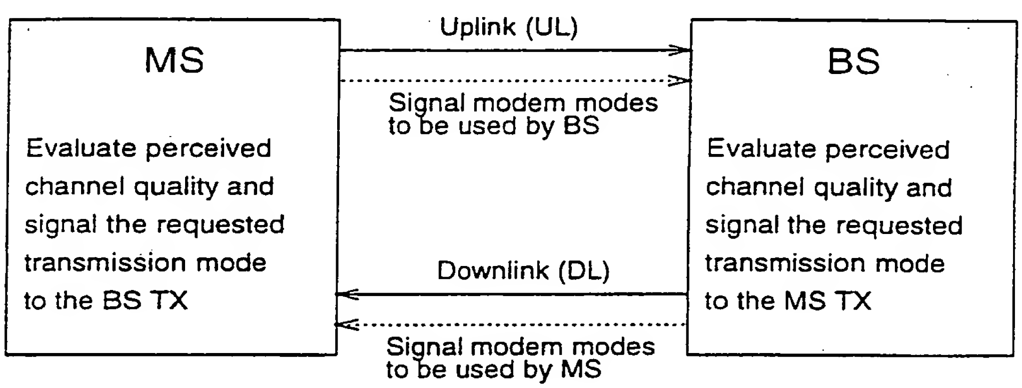


Figure 1(b)

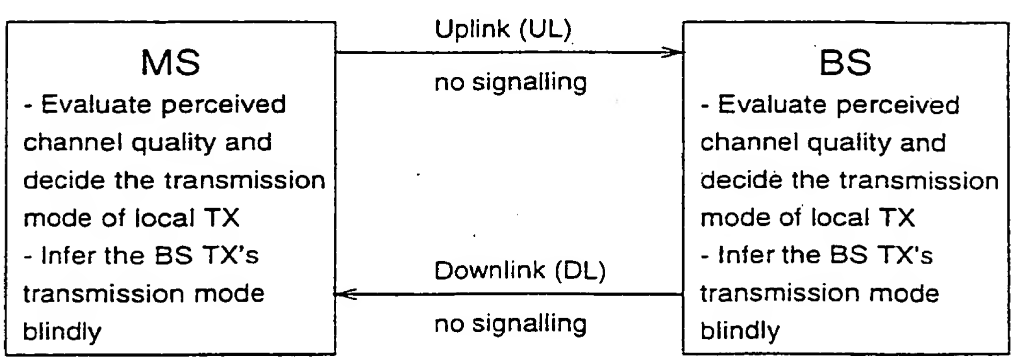


Figure 1(c)

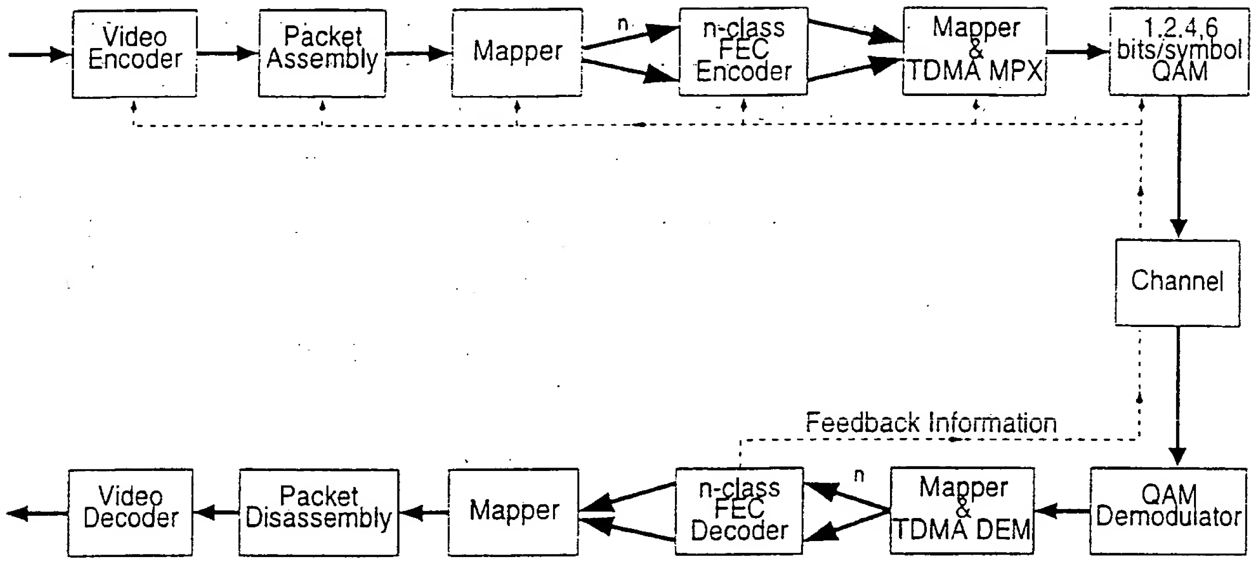


Figure 2

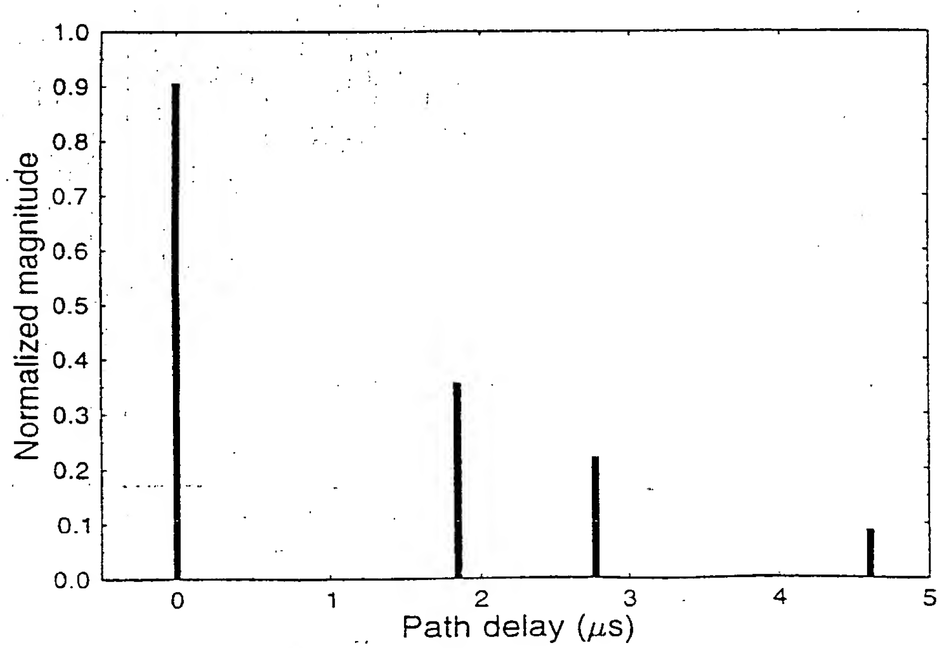


Figure 3

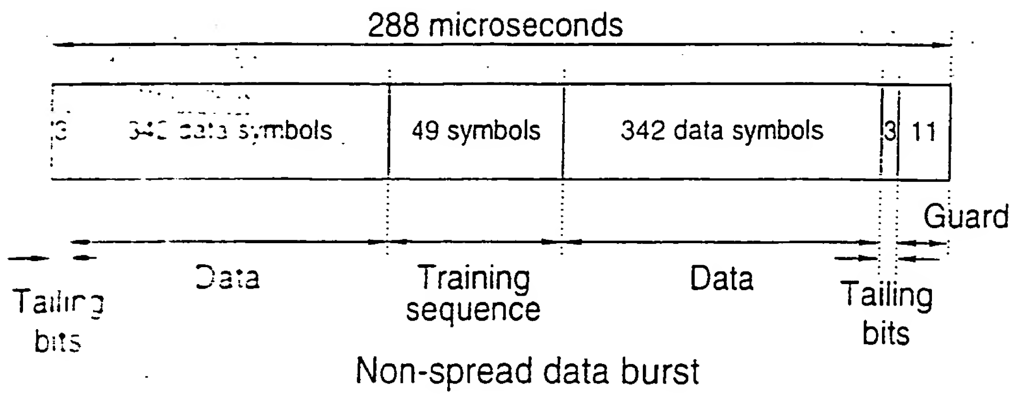


Figure 4

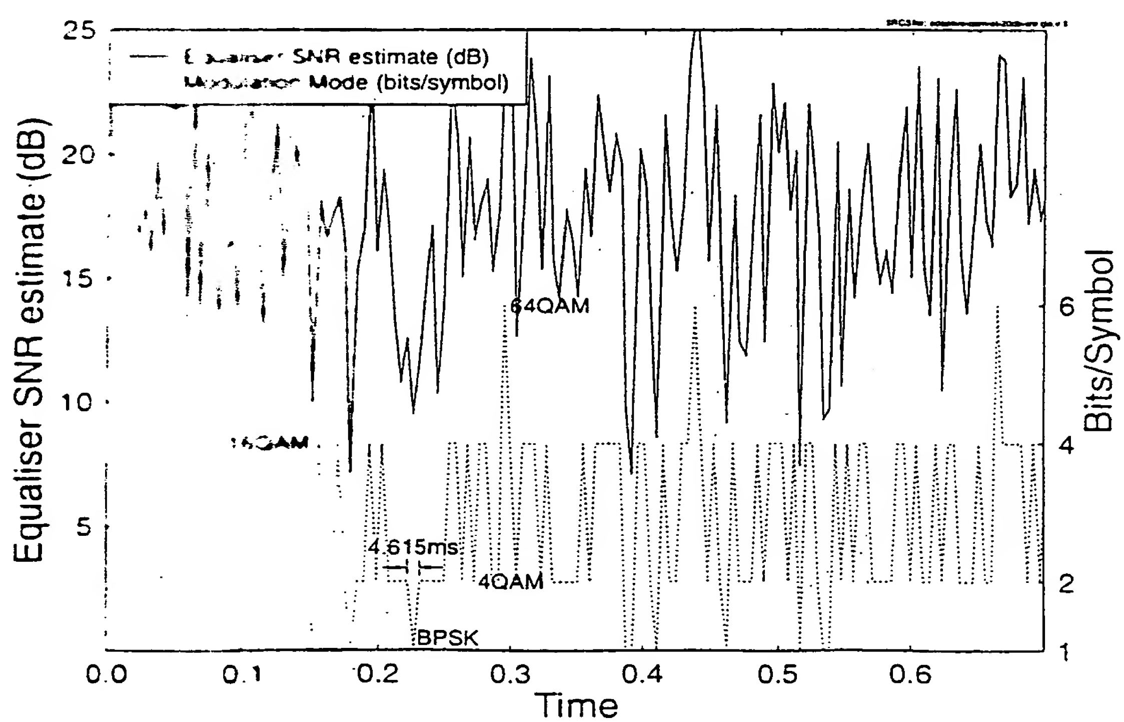


Figure 5

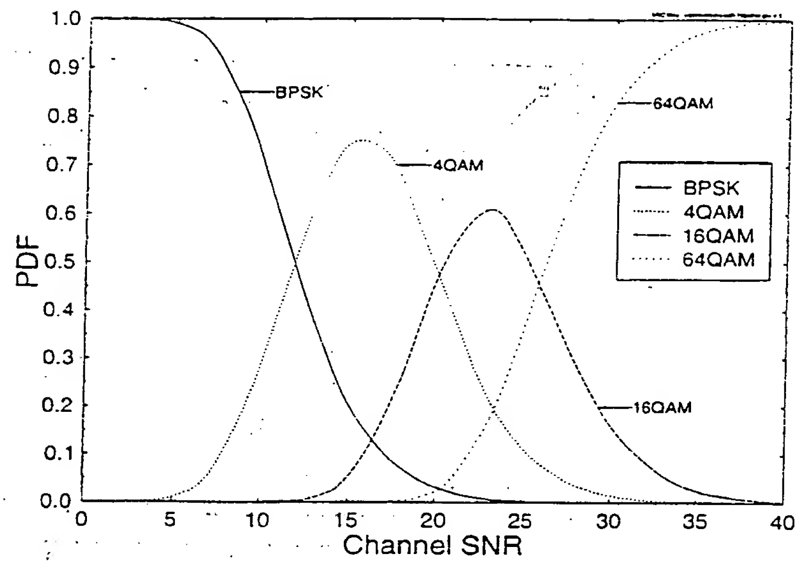


Figure 6

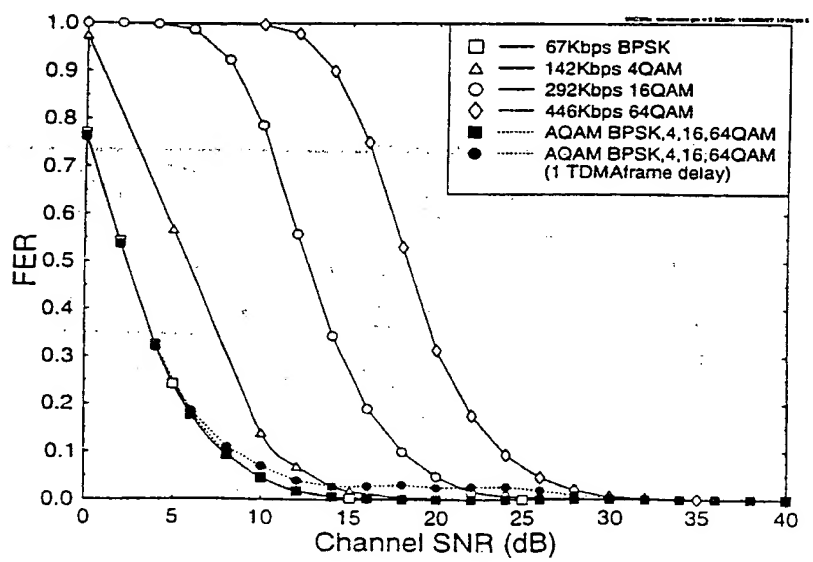


Figure 7

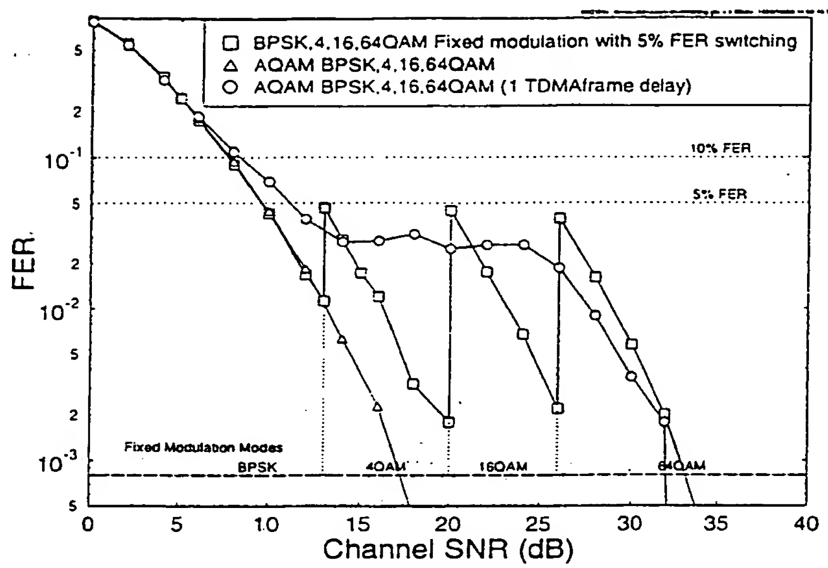


Figure 8

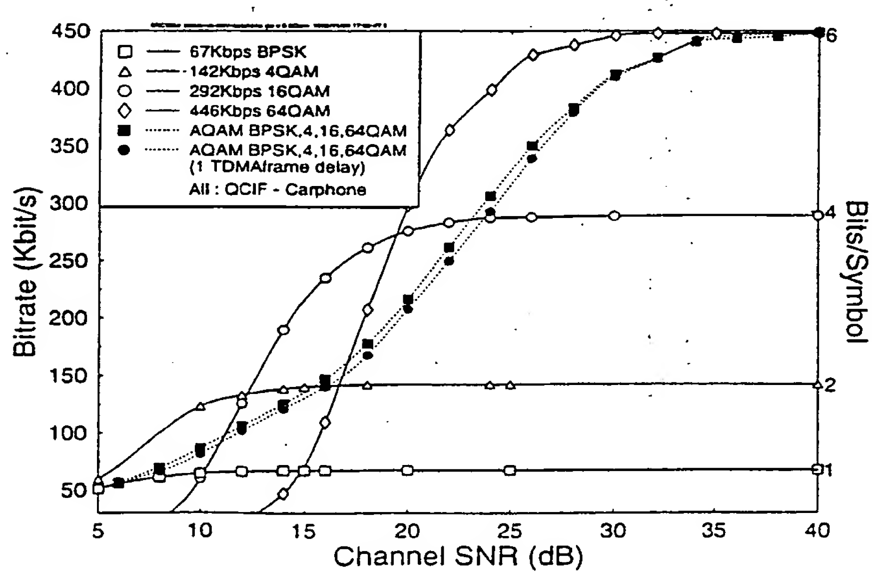


Figure 9

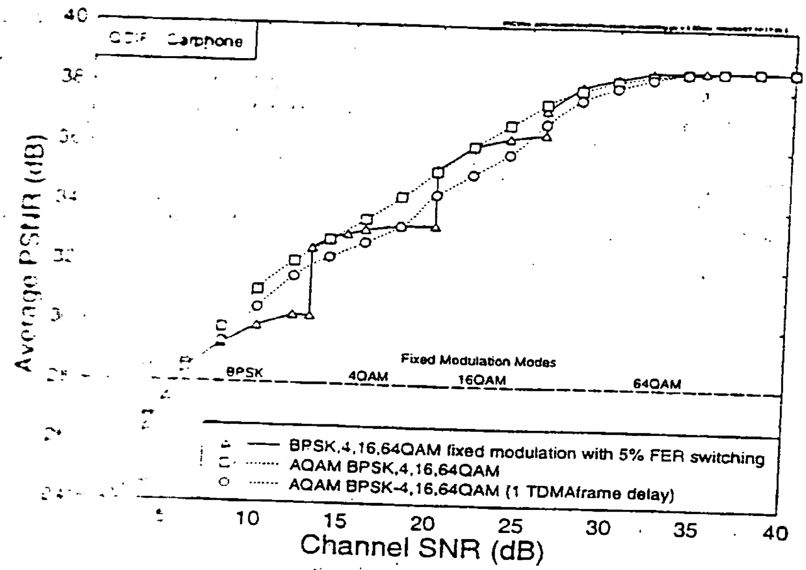


Figure 10

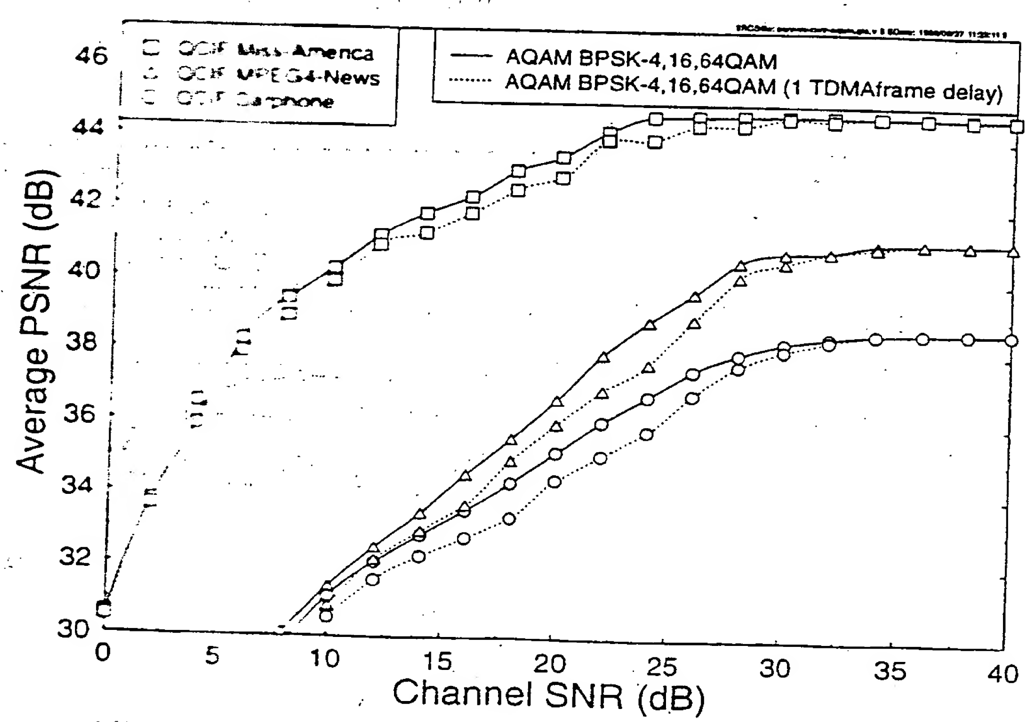


Figure 11

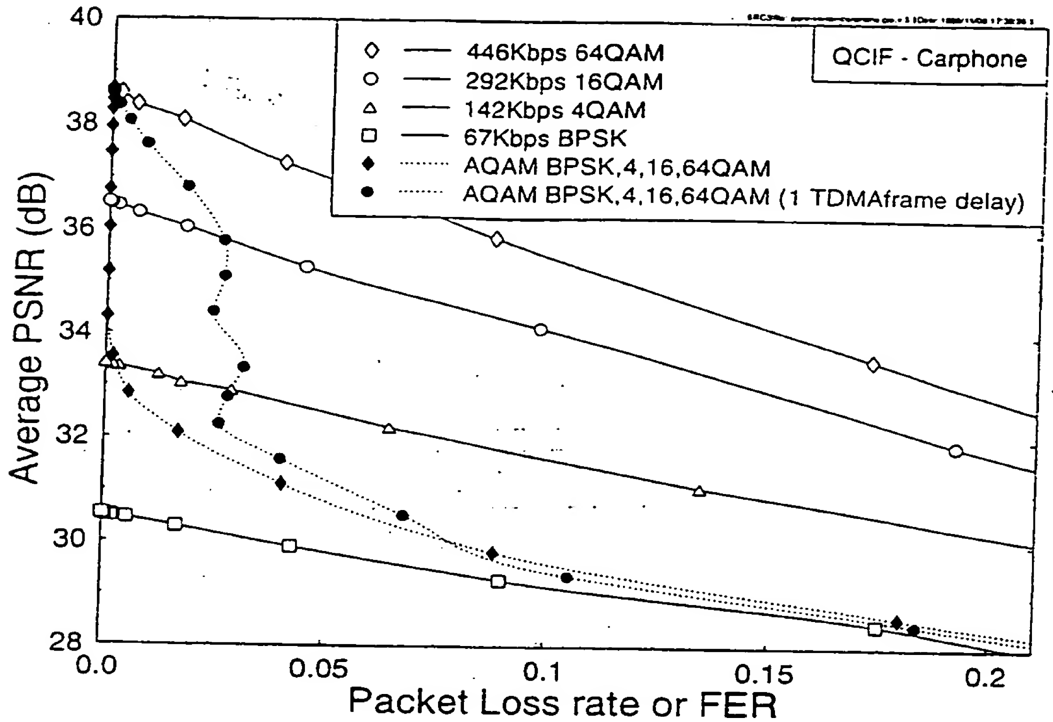


Figure 12

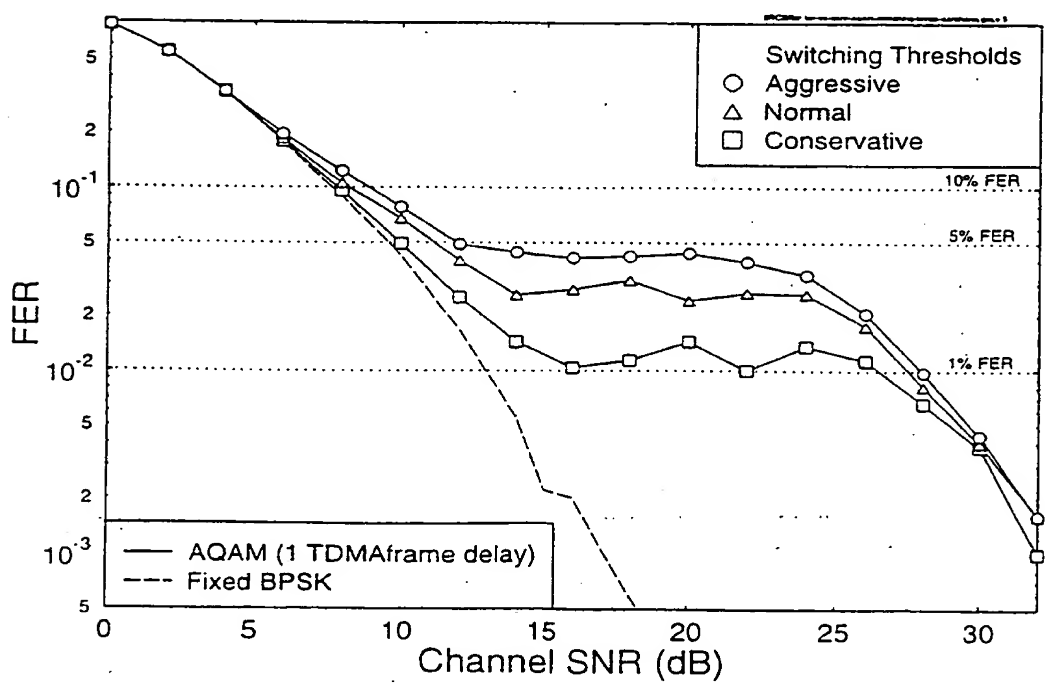
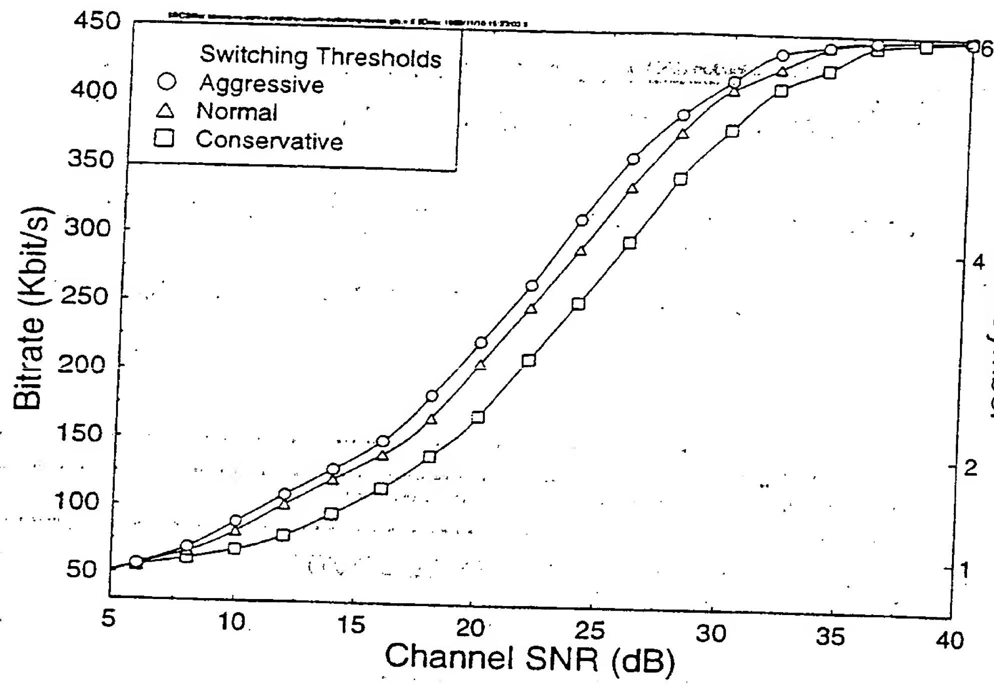


Figure 13



Switching Thresholds
Aggressive
Normal
Conservative
Bitrate (Kbit/s)
Bits/Symbol
Channel SNR (dB)
450
400
350
300
250
200
150
100
50
5
10
15
20
25
30
35
40
6
4
2
1

Figure 14

INTERNATIONAL SEARCH REPORT

Inten nal Application No
PCT/GB 00/00017

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04B1/10

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 98 51111 A (KONINKL PHILIPS ELECTRONICS NV ;PHILIPS AB (SE)) 12 November 1998 (1998-11-12) page 14, line 30 -page 16, line 31; figures 1,8 claims 6,14 page 5, line 30 -page 6, line 14 abstract ---	1,7, 12-14, 16-18, 20,21

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
17 February 2000	28/02/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Kolbe, W

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	OTSUKI S ET AL: "PERFORMANCE OF MODULATION-LEVEL-CONTROLLED ADAPTIVE MODULATION SYSTEMS" ELECTRONICS & COMMUNICATIONS IN JAPAN, PART I - COMMUNICATIONS, US, SCRIPTA TECHNICA. NEW YORK, vol. 79, no. 7, 1 July 1996 (1996-07-01), pages 81-93, XP000696376 ISSN: 8756-6621 page 82, right-hand column, line 22 -page 84, left-hand column, line 21; figures 1,2	1,16
Y	SEIICHI SAMPEI ET AL: "ADAPTIVE MODULATION/TDMA WITH A BDDFE FOR 2 MBIT/S MULTI-MEDIA WIRELESS COMMUNICATION SYTEMS" PROCEEDINGS OF THE VEHICULAR TECHNOLOGY CONFERENCE, US, NEW YORK, IEEE, vol. CONF. 45, 1995, pages 311-315, XP000550185 ISBN: 0-7803-2743-8 page 311, right-hand column, line 6 -page 312, right-hand column, line 4; figure 1	1,16
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Information on patent family members

International Application No
PCT/GB 00/00017

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